

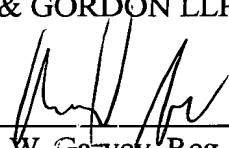
REMARKS

If there are any fees resulting from this communication, please charge the same to our Deposit Account No. 16-0820, our Order No.33234.

Respectfully submitted,

PEARNE & GORDON LLP

By

  
Michael W. Garvey, Reg. No. 35878

526 Superior Avenue East, Suite 1200  
Cleveland, Ohio 44114-1484  
(216) 579-1700

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE SPECIFICATION:

For each occurrence of the title of the invention, including on page 1, delete  
“TRANSIENT” and insert --MOMENTARY--

The paragraph beginning on page 1, line 8, should be replaced with the following rewritten paragraph:

--This invention relates to a method for identifying a [transient] momentary acoustic scene, an application of said method in conjunction with hearing devices, as well as a hearing device.--

The paragraph beginning on page 2, line 6, should be replaced with the following rewritten paragraph:

-- For the extraction of characteristics in audio signals, J.M. Kates in his article titled “Classification of Background Noises for Hearing-Aid Applications” (1995; Journal of the Acoustical Society of America 97(1), pp 461-469), suggested an analysis of time-related sound-level fluctuations and of the sound spectrum. On its part, the European patent EP-B1-0 732 036 proposed an analysis of the amplitude histogram for obtaining the same result. Finally, the extraction of characteristics has been investigated and implemented based on an analysis of different modulation frequencies. In this connection, reference is made to the two papers by Ostendorf et al titled “Empirical Classification of Different Acoustic Signals and of Speech by Means of a Modulation-Frequency Analysis” (1997, DAGA 97, pp 608-609), and “Classification of Acoustic Signals Based on the Analysis of Modulation Spectra for Application in Digital Hearing Aids” (1998, DAGA 98, pp 402-403). A similar approach is described in an article by Edwards et al titled “Signal-processing algorithms for a new software-based, digital hearing device” (1998, The Hearing Journal 51, pp 44-52). Other possible characteristics include the sound-level transmission itself or the zero-passage rate as described for instance in the [article] book by H.L. Hirsch, titled “Statistical Signal Characterization” (Artech House 1992). It is evident that the characteristics used to date for the analysis of audio signals are strictly based on system-specific parameters.--

The paragraph beginning on page 3, line 3, should be replaced with the following rewritten paragraph:

--It is fundamentally possible to use prior-art pattern identification methods for sound classification purposes. Particularly suitable pattern-recognition systems are the so-called [ranging devices] distance classifiers, Bayes classifiers, fuzzy-logic systems and neural networks. Details for the first two of the methods mentioned are contained in the publication titled "Pattern Classification and Scene Analysis" by Richard O. Duda and Peter E. Hart (John Wiley & Sons, 1973). For information on neural networks, reference is made to the treatise by Christopher M. Bishop, titled "Neural Networks for Pattern Recognition" (1995, Oxford University Press). Reference is also made to the following publications: Ostendorf et al, "Classification of Acoustic Signals Based on the Analysis of Modulation Spectra for Application in Digital Hearing Aids" (Zeitschrift für Audiologie (Journal of Audiology), pp 148-150); F. Feldbusch, "Sound Recognition Using Neural Networks" (1998, Journal of Audiology, pp 30-36); European patent application, publication number EP-A1-0 814 636; and US patent, publication number US-5,604,812. Yet all of the pattern-recognition methods mentioned are deficient in one respect in that they merely model static properties of the sound categories of interest.--

The paragraph beginning on page 3, line 19, should be replaced with the following rewritten paragraph:

--One shortcoming of these earlier sound-classification methods, involving characteristics extraction and pattern recognition, lies in the fact that, although unambiguous and solid identification of voice signals is basically possible, a number of different acoustic situations cannot be satisfactorily classified, or not at all. While these earlier methods permit a distinction between pure voice or speech signals and "non-speech" sounds, meaning all other acoustic surroundings, that is not enough for selecting an optimal hearing program for a [transient] momentary acoustic situation. It follows that the number of possible hearing programs is limited to those two automatically recognizable acoustic situations or the hearing-aid wearer himself has to recognize the acoustic situations that are not covered and manually select the appropriate hearing program.--

The paragraph beginning on page 4, line 7, should be replaced with the following rewritten paragraph:

--It is therefore the objective of this invention to introduce first of all a method for identifying a [transient] momentary acoustic scene which compared to prior-art methods is substantially more reliable and more precise.--

The paragraph beginning on page 4, line 15, should be replaced with the following rewritten paragraph:

--The invention is based on an extraction of signal characteristics, a subsequent separation of different sound-sources as well as an identification of different sounds. In lieu of or in addition to system-specific characteristics, auditory characteristics are taken into account in the signal analysis for the extraction of characteristic features. These auditory characteristics are identified by means of Auditory Scene Analysis (ASA) techniques. In another form of implementation of the method per this invention, the characteristics are subjected to a context-free or a context-sensitive grouping process by applying the [g]Gestalt principles. The actual identification and classification of the audio signals derived from the extracted characteristics is preferably performed using Hidden Markov Models (HMM). One advantage of this invention is the fact that it allows for a large number of identifiable sound categories and thus a greater number of hearing programs which results in enhanced sound classification and correspondingly greater comfort for the user of the hearing device.--

The paragraph beginning on page 5, line 14, should be replaced with the following rewritten paragraph:

--The hearing device 1 incorporates in conventional fashion two electro-acoustic converters 2a, 2b and 6, these being one of several microphone 2a, 2b and a speaker 6, also referred to as a receiver. A main component of a hearing device 1 is a transmission unit 4 in which, in the case of a hearing aid, signal modification takes place in adaptation to the requirements of the user of the hearing device 1. However, the operations performed in the transmission unit 4 are not only a function of the nature of a specific purpose of the hearing device 1 but are also, and especially, a function of the momentary acoustic scene. There have already been hearing aids on the market where the

wearer can manually switch between different hearing programs tailored to specific acoustic situations. There also exist hearing aids capable of automatically recognizing the acoustic [scene] environment. In that connection, reference is again made to the European patents EP-B[!]<sup>1</sup>-0 732 036 and EP-A1 814 636 and to the US patent 5,604,812, as well as to the "Claro Autoselect" brochure by Phonak Hearing Systems (28148 (GB) /0300, 1999).--

The paragraph beginning on page 7, line 6, should be replaced with the following rewritten paragraph:

--It is essentially based on the extraction of characteristic features from an acoustic signal during an extraction phase, whereby, in lieu of or in addition to the system-specific characteristics - - such as the above-mentioned zero-passage rates, time-related sound-level fluctuations, different modulation frequencies, the sound level itself, the spectral peak, the amplitude distribution etc. -- auditory characteristics as well are employed. These auditory characteristics are determined by means of an Auditory Scene Analysis (ASA) and include in particular the [volume] loudness, the spectral pattern (timbre), the harmonic structure (pitch), common build-up and decay times (on-/offsets), coherent amplitude modulations, coherent frequency modulations, coherent frequency transitions, binaural effects etc. Detailed descriptions of Auditory Scene Analysis can be found for instance in the articles by A. Bregman, "Auditory Scene Analysis" (MIT Press, 1990) and W.A. Yost, "Fundamentals of Hearing - An Introduction" (Academic Press, 1977). The individual auditory characteristics are described, inter alia, by A. Yost and S. Sheft in "Auditory Perception" (published in "Human Psychophysics" by W.A. Yost, A.N. Popper and R.R. Fay, Springer 1993), by W.M. Hartmann in "Pitch, Periodicity, and Auditory Organization" (Journal of the Acoustical society of America, 100 (6), pp 3491-3502, 1996), and by D.K. Mellinger and B.M. Mont-Reynaud in "Scene Analysis" (published in "Auditory Computation" by H.L. Hawkins, T.A. McMullen, A.N. Popper and R.R. Fay, Springer 1996).--

The paragraph beginning on page 8, line 9, should be replaced with the following rewritten paragraph:

--Another form of implementation of the method according to this invention additionally provides for a grouping of the characteristics in the signal analyzer 7 by means of [g]Gestalt analysis. This process applies the principles of the [g]Gestalt theory, by which such qualitative properties as

continuity, proximity, similarity, common destiny, unity, good constancy and others are examined, to the auditory and perhaps system-specific characteristics for the creation of auditory objects. This grouping – and, for that matter, the extraction of characteristics in the extraction phase – can take place in context-free fashion, i.e. without any enhancement by additional knowledge (so-called “primitive” grouping), or in context-sensitive fashion in the sense of human auditory perception employing additional information or hypotheses regarding the signal content (so-called [design] schema-based” grouping). This means that the contextual grouping is adapted to any given acoustic situation. For a detailed explanation of the principles of the [g]Gestalt theory and of the grouping process employing [g]Gestalt analysis, substitutional reference is made to the publications titled “Perception Psychology” by E.B. Goldstein (Spektrum Akademischer Verlag, 1997), “Neural Fundamentals of Gestalt Perception” by A.K. Engel and W. Singer (Spektrum der Wissenschaft, 1998, pp 66-73), and “Auditory Scene Analysis” by A. Bregman (MIT Press, 1990).--

The paragraph beginning on page 9, line 4, should be replaced with the following rewritten paragraph:

--The advantage of applying this grouping process lies in the fact that it allows further differentiation of the characteristics of the input signals. In particular, signal segments are identifiable which originate in different sound-sources. The extracted characteristics can thus be mapped to specific individual sound sources, providing additional information on these sources and, hence, on the current [transient] auditory scene.--